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NOISE MEASUREMENTS AND THE DB.(U)
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(6) NOISE MEASUREMENTS AND THE dB.

by

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SUMMARY

This Memorandum presents the author's opinion that the conventional use of the dB in studies of acoustic noise can obscure and complicate simple relationships and lead to the possibility of error because of misunderstandings of the nature of logarithms and because the order of the numbers involved is concealed by the compressed scale. Examples of errors and inaccuracies are given, and some cases where consideration of noise in terms of rms pressure would be simpler, or would lead to better understanding of the processes involved, are discussed. It is suggested that, although the dB may be a convenient unit for many purposes, more emphasis on the physical nature of noise would help in the interpretation of phenomena and prevent some of the confusions that arise, especially in dealing with the general public.

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## 1 INTRODUCTION

One of the first tasks that faces the newcomer to the study of acoustic noise is learning to understand and manipulate the decibel (hereinafter written 'dB'). Once this task is thoroughly accomplished, and we begin even to think in dB, we may be tempted to regard ourselves as part of an elite band of cognoscenti, entitled to despise the ignorance of the common herd. Whether the understanding of noise gained through our hard-won knowledge of the dB is worth the trouble of learning is a question that may intrude occasionally - at least, it has on me. But the urge to suppress it is powerful. We have struggled to gain the art of dealing with the dB, we understand it, it is part of the mystique of acoustics. Moreover, if the experts over the years have apparently derived inestimable benefit from using the dB, who are we, newcomers to the subject, to doubt its utility?

Well, the fact that we are going against expert opinion need not deter us. Experts can be mistaken; even in science, today's heresy may become tomorrow's orthodoxy.

But is it worth the effort? Even if we could show conclusively that the use of the dB is prejudicial to the understanding of noise, could anything be done about it? The dB has been in use for so long, there is so much elaborate measuring equipment calibrated in dB, there are so many national and international standards expressed in dB, that to eliminate its use would be a very long process, and a very expensive one. But not impossible. There must be many more instruments in Britain calibrated in Imperial units than were ever calibrated in dB, and they are rapidly being phased out of use. So, if it were shown to be eminently desirable to get rid of the dB totally, it could be done.

In fact, I don't advocate the abolition of the dB, all I am suggesting in this paper is that we should adopt a more critical approach to its use. To anticipate my argument a little, I believe, and shall attempt to show in this paper, that the dB can obscure and complicate simple relationships, and lead to the possibility of error and misunderstanding because the order of pressure involved is suppressed and the arithmetic required for some calculations is cumbersome and of a kind where small slips can lead to large errors. I think that, though it may be too late in the day to expect acousticians to talk and think in other terms, a little more stress on the fact that we are concerned with pressure oscillations would encourage clearer thought and benefit newcomers to the subject - and the general public - by allowing them to understand without

having to cope with logarithms. I would hazard a guess that not more than 20 per cent of the population of this country has any understanding at all of the nature of logarithms, and, in these days of modern maths and pocket calculators, even fewer have any practical experience of their use.

This paper then, is a short and somewhat discursive enquiry into some aspects of the use of the dB in noise studies. I shall discuss the meaning of the dB in general and in relation to various applications, methods of manipulating the dB, and some cases where the use of the dB may hide important physical relationships

Taken together, I believe that these considerations demonstrate that the dB ought to be used with greater care than it commonly is, and that there are many cases where it would be more efficient to work in pressure directly, either from the point of view of reducing the quantity of calculations required or of understanding the processes involved.

## 2 THE dB AND NOISE MEASUREMENTS

### 2.1 Why do we use the dB?

The reasons for using dB in noise measurements have been stated many times by many authors. I reproduce one such statement, that given by Broch[1] in a handbook published by Messrs Bruel and Kjaer. Broch says: "The weakest sound pressure to be detected by the 'average' person at 1000 Hz has been found to be 0.0002 μbar ($2 \times 10^{-5}$ N/m$^2$)*. On the other hand, the largest sound pressure perceived without pain is of the order of 1000 μbar, *ie* the scale of sound pressures covers a dynamic range of 1:1000000! The use of sound pressures in μbar directly as a measure of sound measurements is therefore not too convenient. Also ... the hearing mechanism responds to changes in sound pressure in a relative rather than in an absolute way." Broch goes on to define the dB thus: "The decibel is defined as ten times the logarithm to the base 10 of the ratio between two quantities of power. As the sound power is related to the square of the sound pressure, a convenient scale for sound (noise) measurements is defined as

$$\text{sound pressure level} = 10 \log(p^2/p_0^2) = 20 \log(p/p_0) \text{ dB ."}$$

* The units used for pressure have been highly variable over recent years. The rms pressure at the nominal threshold of hearing has been variously defined as 0.0002 μbar, $2 \times 10^{-4}$ dyne/cm$^2$, $2 \times 10^{-5}$ Newton/m$^2$ and 20 μPa (micro Pascal). The last is now the accepted form, but in this paper the forms originally used in quotations will be retained.

Thus, there are two reasons put forward for using a logarithmic scale:

(1) the range of pressure to be covered is too large to be dealt with easily using linear pressure units, and

(2) the human hearing mechanism responds to audio stimuli in a logarithmic manner.

The first reason is made to sound even more striking by some authors who talk about intensity instead of sound pressure, which they define, often merely by implication, as proportional to pressure squared, thus obtaining a ratio of $10^{12}$ instead of $10^6$. Broch is more cautious, he says merely that power is 'related' to the square of the sound pressure, and as he says (see above) the dB is basically a unit of power ratio. We shall return to this point later (section 2.2).

2.1.1 Range of perception

It is said to be necessary to compress the sound measurement scale because of the large range of perception. Consider the following argument. The smallest dimension which can be perceived by the human eye is of the order of 0.2 mm. On the other hand, distances as great as several km can be easily observed. That is, the range of distances perceptible by the human eye is of the order of $1:10^6$ or, if we think of the areas which can be seen, of the order of $1:10^{12}$. The use of metres directly for measurements of distance is therefore not very convenient. We therefore define the dB for length or area as ten times the common logarithm of the ratio of the area to the area generated by the smallest perceptible distance ($d_0$ = 0.2 mm), that is

$$\text{the distance level} = 10 \log(d^2/d_0^2) = 20 \log(d/d_0) \quad .$$

Using this reference length, 1 mm becomes 14 dB, 1 m becomes 74 dB, and 1 km becomes 134 dB re 0.2 mm.

One has only to state it in these terms to see the illogicality of the argument that a logarithmic scale is needed because of the range of values involved. By this argument, if we need the dB for sound, we also need it for every other quantity that is perceptible or measurable - the human is only one form of measuring instrument. But we know this is not so. The average man, and even more the average scientist, can manage perfectly well with linear units, using those appropriate to the size of the thing he is measuring - in the metric system he merely selects the most suitable prefix for the unit. Is the range between 20 μPa and 20 Pa really too much to cope with along the same lines?

After I had written this paragraph, I came across an author[2] who had used very nearly the same analogy, and deduced from it that the dB was necessary for sound measurements "... because we are just as interested in measuring small pressures as large ones and it would be impossible to construct a single linear rule to measure inches and miles." But, when miles are in question, no-one is interested in inches, neither when sound pressures of the order of Pa are in question are we concerned with μPa. And if we were, we should need an instrument with a dynamic range of 120 dB to measure both without changing the scale. If such instruments exist for any applications, these certainly do not include acoustics where the dynamic range is commonly 60 or even 40 dB.

2.1.2 Human perception and the dB

Broch's statement that "the hearing mechanism responds to changes in sound pressure in a relative manner" is explained later by reference to the Weber-Fechner law, which he states as follows: "The Weber-Fechner law states that the change in response to a certain change in excitation is inversely proportional to the absolute excitation before the change. Mathematically this can be written $dR/dE = k/E$ ."

Integrating this equation, we have $R = k \ln E$ , which is the way the law is most commonly stated. The physiological sensation produced by a stimulus is proportional to the logarithm of the stimulus.

Two points should be noted about the Weber-Fechner law. First, it affects acoustics only inasfar as acoustics is concerned with man's response to sound - it can have no bearing on the units used in many acoustical studies, for example, the response of structures to air pressure vibrations. To use units for a phenomenon which has many effects based solely on its effects on one receptor (the ear) does not seem to accord well with the best interests of science.

Secondly, the law relates not only to sound but also to all stimuli which produce human sensations. Hence, if the law demands the use of the dB for sound, dB units should also be used for all other such stimuli. Not so very long ago one could have said with confidence that no such other units existed; but of recent years attempts have been made to introduce a dB scale for vibration[3] and dB have also been introduced into optics and other sensory topics. In the case of vibration, since the full range of human perception is at the most about 10000, the first reason for using the dB for noise cannot apply, and the case must rest solely on the Weber-Fechner law. In fact, the real reason for this introduction is probably that the inventors of the vibration dB were

accustomed to using dB in noise, and extended it to vibration by analogy. This must also be the reason why some writers on noise (*eg* Barry[5]) have extended the dB to other factors such as time involved in their experiments - not always with perfect consistency (see section 2.3).

## 2.2 Is the dB used correctly in noise studies?

As Broch[1] says (see section 2.1), the dB is a unit for power ratio, and is so defined in technical dictionaries, for example, Chambers[4] gives for the bel: A non-dimensional unit, ten times the size of the more frequently used decibel, expressing ratio of power units ($P_1$ and $P_2$)

$$N = \log_{10}(P_1/P_2) \text{ bels} \quad .$$

(One wonders, incidentally, how the dB has managed to survive the introduction of SI units, which eliminate the prefix deci! We ought, surely, to use either bels or millibels - but then we should no longer have the advantage of having to deal generally only with smallish integers!). The ISO terminology for mechanical vibration and shock[6] extends the definition somewhat, speaking of the ratio of 'power-like quantities', and includes among quantities that qualify as power-like, sound-pressure squared, sound intensity and sound energy density. In fact, of course, the dB as usually used in acoustics is related to the sound pressure, simply because that is what microphones respond to.

It is true that, in a given situation and for a given frequency, the power is proportional to the pressure squared, but changing the frequency or the conditions will change the power for the same sound pressure. If we really require the power, something else must be measured as well as pressure, pressure gradient or particle velocity for instance.

Thus the dB as used in acoustics is not always strictly comparable with the dB of communications theory or electrical engineering. The distinction is not generally important, so long as workers in noise remember that what they are measuring is usually simply pressure squared.

## 2.3 Advantages and dangers of using dB in sound measurements

The cynic may feel that there are two primary advantages in the use of dB in acoustics:

(1) it is not easily understood by the laity, so the acoustician's mystique is preserved,

(2) it gives an air of accuracy to a very inexact science: '94 ± 6 dB re 20 μPa' sounds much more accurate than '1 Pa + 1 or −0.5 Pa'!

The real advantages are, perhaps

(3) a scale of manageable (mostly 2-digit) numbers can be used for all normal measurements,

(4) ratios of pressures or of the powers associated with the pressures are expressed by the same number, since

$$20 \log(p/p_0) = 10 \log(p^2/p_0^2) \quad ,$$

(5) amplification, attenuation etc can be obtained by addition and subtraction - simpler arithmetical processes than multiplication and division, and the shapes of spectra and frequency responses are independent of the level - but this could be achieved, as it is in other disciplines, simply by plotting pressure or pressure ratio on a log scale.

The dangers are associated with the advantages:

(1) the general public does not understand the dB, and cannot be expected to,

(2) inaccuracies and variations of an order which would make workers in almost any other field abandon their measurements as meaningless are accepted in acoustics because,

(3) the real variations in the order of sound pressures are disguised by the compressed scale. As a speaker on 'acoustic mythology' said[7]:

"The glib use of the decibel often leads to a misplaced sense of proportion. Acoustics involves an immense range of magnitudes and the decibel scale provides a system of measurement that corresponds with the way the human hearing system behaves. It is easy to become so confused by the decibel scale that the size of the numbers being dealt with is forgotten - which can lead to expensive mistakes."

The glib use of the dB can also lead to loss of accuracy. People working on noise do not say "we doubled the sound power" or "the sound pressure varied inversely with frequency". They say instead "we increased the level by 3 decibel" or "the level decreased by 6 decibel per octave". The first forms are straightforward expressions which can be understood by anyone with elementary mathematics and tell you a lot about the physical conditions. The second forms

are not only more obscure, they are not even precise equivalents, since 3 dB actually represents a power ratio of 1.995. The difference hardly matters in experimental work; but it is incorrect - and very distasteful to a mathematician - to see exact theoretical forms described, inexactly, in terms of dB.

(4) Confusion between pressure and pressure squared (so-called power), and between either and ratio is common.

The first kind of confusion is worst when the dB is used with quantities other than sound pressure. For instance, one author[5] wished to obtain a linear equation for a plot of dB against the logarithm of time. He therefore used a symbol $t$ which he defined as 'the time expressed in dB re 1 hour'. Close investigation showed that $t$ was ten times the log of the time in hours - from the definition it might as easily have been 20 times. In any case, such a use of dB is quite indefensible - time, even time squared, is nothing like power.

The second kind of confusion can arise between the dB as a pure ratio, for instance as an amplification or attenuation factor, and dB as a ratio to a reference pressure, in which case it represents a quantity of sound. Qualifications such as A, B or lin imply the second case, for instance, 94 dB(A) means the level of a sound which after passing through an A-weighting filter has an rms pressure 94 dB above 20 μPa. That is, the A-weighted rms pressure is $20\ \mu\text{Pa} \times 10^{94/20} = 1.002$ Pa.

If we attenuate such a noise by 20 dB, we multiply the rms pressure by $10^{-20/20} = 0.1$ , and we then have an rms pressure of 0.1 Pa, corresponding to 74 dB(A). But some people say they attenuate a noise by 20 dB(A). Such a phrase can have no meaning, unless we read 'attenuate' in its non-scientific meaning of 'reduce'. And then it means that the noise is reduced by removing 20 dB(A) from 94 dB(A). As a simple sum shows, this would leave not 74 dB(A) but 93.9999998 dB(A) - not much of an attenuation!

(5) Since amplification and attenuation are obtained by adding or subtracting dB, and since one measures, apparently, microphone outputs in dB, it is easy for the beginner to forget that dB are in fact ratios, and that the unprocessed signals coming out of microphones are not like the dB readings.

A young experimenter of my acquaintance, after several months work on acoustics, attempted to measure the attenuation afforded by a hearing protector by measuring the *difference* between the unprocessed outputs from microphones positioned inside and outside the protector! Another, trying to measure the

distortion in a system, recorded the levels of the fundamental and harmonics in dB. He then inserted the figures he obtained in the well-known formula - distortion is the square root of the sum of the squares of the harmonic amplitudes divided by the fundamental amplitude - and found that he had nearly 100 per cent distortion, although a glance at his cathode ray oscilloscope showed that the reproduction was nearly perfect.

Such kinds of confusion would be at least reduced if those familiar with the subject would avoid the use of such words as 'add' and 'subtract', 'sum' and 'difference', unqualified, in connection with dB. Some handbooks do not help by such confusing statements as the following (I quote literally from a source that shall be nameless):

"If $S_1$ and $S_2$ are two sound pressures, the difference between them is

$$db = 20 \log(S_1/S_2)" \quad .$$

(6) Another cause of confusion is the habit that has grown up, especially in relation to impulse noise, of marking dB as scales on the recordings of microphone outputs. Fig 1a&b show two examples of how this is sometimes done. Example a is just defensible, as indicating the level of the peak in relation to 20 μPa. If the reference is omitted, however, it would seem to imply that the peak, in this case, is at 140 dB referred to the zero line which represents ambient pressure. This would often mean that the negative pressures went well below zero pressure, which is clearly absurd. Example b is quite indefensible. It implies that the range from zero pressure change to the peak is 140 dB. But zero pressure change is $-\infty$dB, referred to any pressure whatever, so the range from zero change to any pressure whatever is $\infty$dB! The dB scale re 20 μPa rms (which is 28.28 μPa peak) would have to look as shown in Fig 1c, with the range from $\infty$ to 120 dB compressed into 10 per cent of the scale and the entire range - from $-\infty$ to 100 dB squashed into the bottom 1 per cent of the scale. Presumably it should also be repeated on the negative pressure side, as shown. Such a scale looks very strange, and there could be argument as to whether it is correct to denote a transient pressure in this way.

Two reasons have been advanced for marking transient pressure in the ways shown in Fig 1a&b. The first reason is that calibrations are performed using instruments calibrated in dB. But it is only necessary to do a simple sum once to determine the peak pressure produced by a calibrator, for example, 124 dB re 20 μPa gives peak pressures of ambient ±44.83 Pa. The scale may then easily be marked off as shown in Fig 1d.

For the second reason, it is said that the acoustics world can only understand noise when it is described in dB. Unfortunately, this is probably true; but scales such as those shown in Fig 1d, annotated, if necessary to give the peak pressure in dB referred to 20 μPa should be readily understood even by the most hidebound acoustician.

## 3 MANIPULATING dB

### 3.1 Adding up acoustic energy, and noise dose calculations

The major advantage, in practice, of using dB is the ease with which multiplication and division can be accomplished. But you never get something for nothing, even in mathematics, and this simplicity entails the converse, that real addition and subtraction are difficult. This section will therefore deal mainly with problems relating to addition.

The first time that the difficulty of totting up dB levels strikes the newcomer to acoustics, is probably when he wishes to determine the overall sound pressure level, given the levels ($L_1$, $L_2$, ... dB) in a number of octave or third-octave bands. The well-known formula

$$\text{OASPL} = 10 \log\left[\sum_1^m 10^{L_r/10}\right] \quad \text{or} \quad 10 \log\left[\sum_1^n \text{antilog}(L_r/10)\right] \tag{1}$$

is, of course, not particularly complicated - when you get used to it - but it does not immediately convey much to the mind, nor can it be used without aids, log tables at least, or computer programs or special nomograms.

Again, the formulas for averaging acoustic energy over time

$$\bar{L} = 10 \log \int_0^T 10^{L/10} dt/T \tag{2}$$

and the corresponding summations for discrete intervals, are simple in essence, but can be very wearisome in calculation.

The literature teems with graphs, nomograms and tables for performing this kind of calculation (two examples are shown in Fig 2); and in specifications for permissible noise exposure, for example, much space is taken up by the methods of computation. For example, in the Department of Employment "Code of practice for reducing the exposure of employed persons to noise"[8], large

sections are devoted to adding up the noise energy. First, if the noise level lasts for a shorter period than the 8-hour working day, or there are different levels during the day, one has to calculate the equivalent 8-hour level, by means of a nomogram. Second, with a statistical count, one has to add up the periods in each class and determine the overall level. Third, if one measures a spectrum, one has to ascertain the overall weighted level of the noise. For each of these calculations, nomograms or tables are provided - and it is difficult to see how the novice could manage without them.

And yet, actually, the calculations are very easy, it is only the dB that makes them seem complicated.

All that is really required is to determine the overall mean square or root mean square pressure at a given time (equation (1)), or the mean square pressure over an 8-hour day (equation (2)). Stated in these terms, not only do most of the arithmetic difficulties disappear, but the physical meaning of the process is plain.

Thus, if we measure the rms pressures $p_1$, $p_2$, ... $p_n$ Pa in n bands, the overall rms pressure is $\bar{p}$ Pa,

where
$$\bar{p}^2 = \sum_{1}^{n} p_r^2 \quad . \tag{3}$$

It may be noticed that this equation omits the reference pressure. This is intentional, since the usefulness of the reference pressure is confined to the logarithmic scale - where it is essential. If actual pressures are used there is no need of a reference pressure.

Similarly, in considering the effects of noise on hearing, the noise dose X $(Pa)^2$-hours received in a time T hours, would be defined as

$$X = \int_0^T p_A^2 dt \quad . \tag{4}$$

where $p_A$ Pa is the A-weighted rms pressure at time t .

The maximum daily dose, according to Ref 8, is equivalent to 90 dB(A) for 8 hours, that is a ms pressure of 0.4 $(Pa)^2$ or 0.6325 Pa for 8 hours, or

$$3.2 \ (Pa)^2\text{-hours per day} \quad .$$

Hence, the percentage of the permissible daily dose received in time $T$ is

$$y = 100\,X/3.2 \quad . \tag{5}$$

This is exactly the way that noise dosemeters work, and the form of their read-outs. According to current practice, we take this number $y$, and according to what we are looking for, calculate the equivalent level during the exposure, $L_T$ dB(A), or the equivalent daily level if the noise during the exposure persists for T' hours, $L_{eq}$ dB(A), by means of the equations

$$L_T = 90 + 10\log(8y/100T) \; ; \quad L_{eq} = 90 + 10\log(yT'/100T) \tag{6}$$

or using the nomograms provided in the handbooks.

The $L_{eq}$ is then compared with the maximum figure of 90, and we say either that there is no danger of incurring a hearing loss, or else that we must reduce the dose either by reducing the noise level by so many dB or by reducing the exposure time - and to do this last part of the sum we have to use antilogs or fiddle with the nomogram.

How much easier it would be if we left dB and percentages out of the calculation and simply worked from the value of $X$ in equation (4)!

## 3 EFFECTS OF NOISE SPECTRA ON ATTENUATION

An apparent advantage of using the dB is that we can talk about attenuation in the same terms as amplification. Thus, a noise level may be increased by 10 dB (that is, the energy is multiplied by 10), or attenuated by 10 dB (that is, the energy is divided by a factor 10).

But this sometimes masks the true meaning especially in relation to noise dose. For example if the attenuation given by an ear-protector is 10 dB for one frequency band, and -3 dB for another (by no means an unknown phenomenon), this means that 10 per cent of the energy reaches the ear in the first band, but in the second, the energy is doubled. The importance of these figures will depend on the noise to be attenuated in each band; but it is clear that if the energy outside is approximately the same in each band an improvement of the same number of dB in each band will have far more effect in the second band than in the first.

Fig 3 shows some variations on this theme. Fig 3a shows the further decrease of transmitted energy if the attenuation is increased by 1, 2, 5 or 10 dB. Fig 3b shows the increase of attenuation in dB that would be required to decrease

the proportion of the input energy transmitted by 1, 5, 10, 20 or 50 per cent. It will be seen, *eg*, that an improvement of 10 dB from 0-10 dB will remove 90 per cent of the energy, whereas an improvement from 20-30 dB will decrease the energy transmitted by only about 1 per cent (0.9 per cent). And an increase of attenuation from 0-3 dB will remove 50 per cent of the energy, but to remove the further 50 per cent after an attenuation of 3 dB would require infinite attenuation!

These are very simple and obvious illustrations, but I believe the implications are not always recognized and that enormous effort is sometimes applied to increasing already adequate attenuation, instead of concentrating on those frequencies where a small gain in attenuation would greatly decrease the transmitted energy.

The arguments so far refer to separate bands of noise, but it is also true that the attenuation of a band of noise by a hearing-protector (assuming that the properties of the protector remain constant) will not be independent of the spectrum of noise within the band unless the attenuation at all frequencies within the band is the same. In Ref 9 it was shown by narrow-band measurements that the attenuation of noise by the Mk 4 helmet may vary by as much as 20 dB within a third-octave band. The possible bearing of this kind of variation on the attenuation of a third-octave band of noise was summarised as follows:

"Suppose the ratio of the rms pressure inside to that outside the muff at a frequency $f$ Hz is $t$, equivalent to an attenuation of $(-20 \log t)$ dB, and that the rms amplitude of the sound pressure density at the same frequency is $20X$ μPa, equivalent to $(20 \log X)$ dB. Then the attenuation of noise in the band from $f_1$ Hz to $f_2$ Hz is $A = (-20 \log T)$ dB,

where
$$T^2 = \left[\int_{f_1}^{f_2} (tX)^2 df\right] \Bigg/ \left[\int_{f_1}^{f_2} X^2 df\right] . \tag{7}$$

Some features are obvious from equation (7).

(1) The overall attenuation of noise depends on the distribution of acoustic energy within the band.

(2) If within the band, $t$ is always greater than $t_{min}$ and less than $t_{max}$, then $T$ will also lie between these limits, whatever the noise spectrum.

(3) If all the noise is concentrated in a narrow band about a frequency $f_k$ Hz, say, the effective overall band transission will be the value of $t$ at $f_k$ Hz.

(4) For white noise, $X$ is constant throughout the band, hence, substituting in equation (7), $T_w$ the transmission of white noise is given by

$$T_w^2 = \left[ \int_{f_1}^{f_2} t^2 df \right] \Big/ (f_2 - f_1) \quad . \tag{8}$$

For third-octave bands, the variation in level of pink noise within the band is small, only 1 dB, so that, if $T_p$ is the transmission of pink noise,

$$T_p \simeq T_w \quad ." \tag{9}$$

Fig 4, also taken from Ref 9, illustrates the limits within which the overall attenuation of noise within a band may vary relative to the attenuation of white noise according to the distribution of the noise energy within the band, if there is a step change of attenuation within the band. The maximum and minimum overall attenuations will clearly occur if all the noise is concentrated within the part of the band with maximum or minimum attenuation, as opposed to the case when the noise is more or less evenly spread over the band.

## 4 'EXPONENTIAL' RELATIONSHIPS

### 4.1 General

It is common in acoustics to postulate so-called exponential relationships between noise and some dependent function, using equations of the form known in psycho-acoustic as in other psychological measurements as Stevens' power law:

$$q = A(p/p_r)^n \tag{10}$$

where $q$ is the dependent function, $p$ is the rms sound pressure referred to $p_r$ (generally 20 μPa), and $A$ and $n$ are constants. Such a function has the great advantage that, taking logs, we have

$$\begin{aligned} \log q &= \log A + n \log(p/p_r) \\ &= \log A + (n/20)L \end{aligned} \tag{11}$$

where $L$ is the noise level in dB.

Hence, a plot of $q$ on a log scale against $L$ is a straight line, and the value of the exponent $n$ can be determined from the slope. Thus, if measurements are made in dB, there is a strong incentive to formulate theories based on this equation.

Although equation (10) may represent some functions faithfully, the facile assumption that all relationships in certain fields are of this kind can lead to erroneous conclusions. For example, if there should be a constant added to either side of the equation (10), the plot would no longer be a straight line, but experimental points might still appear to lie on a line from which an exponent could be deduced. Such an exponent, though it represents the form of the function over the experimental range, can lead to a total misunderstanding of the nature of the mechanism involved.

## 4.2 Ultimate TTS in steady noise

It has often been suggested (*eg* in Ref 10) that the ultimate threshold shift in dB, at a given frequency, for exposure to a given kind of noise, is given by an equation of the form

$$TTS_{\infty} = n(L - L_c) \tag{12}$$

where $L$ is the sound level in dB, $L_c$ dB is a so-called critical noise level and $n$ is a constant generally said to be about 1.7. The difficulty that TTS can only be positive and that equation (12) gives rise to negative values for $L < L_c$ is overcome by stating that equation (12) holds only for $L > L_c$. Various formulas to fit results for $L$ only a little greater than $L_c$, where the plots are plainly nonlinear, have been suggested.

I have proposed elsewhere[11] that equation (10) is incorrect for this phenomenon, in that, since we are looking for *changes* in threshold, the left-hand side should be $(q - q_1)$ where $q$ and $q_1$ are the rms pressure thresholds ultimately and initially. Equation (12) then becomes of the form

$$TTS_{\infty} = 20 \log\left\{1 + \text{antilog}\left[n(L - L_c)/20\right]\right\} . \tag{13}$$

This clearly gives positive values for any value of $L$, and I showed that much of the published data fits the assumption that $n = 2$. Such a value of $n$ makes good physical sense, since it suggests that the TTS is due to the absorbed energy.

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## 4.3 Growth of TTS in steady noise

Again, it is customary to attempt to fit straight lines to plots of dB against log time in considering response, for example, the growth of and recovery from TTS. The assumption is that

$$D = A + B \log t \tag{14}$$

where $A$ and $B$ are constants and $D$ dB is the varying level.

If we write equation (14) in pressures taking $p$ for the threshold pressure at time $t$ and $p_1$ the initial threshold in Pa rms, we have

$$(p/p_1) = at^n \tag{15}$$

where $20 \log a = A$ and $20\,n = B$ .

But such equations can hold only if all values of $D$ from $-\infty$ to $\infty$ are possible, which is clearly never the case in acoustics where neither perfect silence nor infinite levels are attainable. Also, the attempt to fit straight lines to plots of this kind is often made without considering whether a relationship of this form is likely. For example, when dealing with TTS, there is no ground for supposing that the shift of the pressure threshold is proportional to a power of the time. Also clearly, one always starts at zero shift and tends to an ultimate asymptotic value in growth, or vice versa in recovery, so that equations like equation (14) can only fit sections of response curves and give, at best, an approximation to the time function.

I have suggested[11], that for TTS the true form is given by the usual exponential functions

$$\left.\begin{array}{lll} \text{in growth} & p - p_1 = (p_2 - p_1)(1 - \exp(-t/T)) \\ \text{in recovery} & p - p_2 = (p_1 - p_2)\exp(-t/T) \end{array}\right\} \tag{16}$$

where $p_1$ and $p_2$ Pa rms are the initial and ultimate pressure thresholds and $T$ is the time constant.

These are simple and easily understood forms. If we turn equation (16) into dB, however, the simplicity is lost, for we then have,

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$$\left.\begin{aligned}
&\text{in growth,} \quad & D &= 20\log(p/p_1) \\
& & &= 20\log\left\{1 + \left[\text{antilog}(D'/20) - 1\right]\left[1 - \exp(-t/T)\right]\right\} \\
&\text{and} \\
&\text{in recovery,} \quad & D &= 20\log(p/p_2) \\
& & &= 20\log\left\{1 + \left[\text{antilog}(D'/20) - 1\right]\exp(-t/T)\right\}
\end{aligned}\right\} \quad (17)$$

where $D'$ is the ultimate shift in growth and the initial shift in recovery. Equations (16), (17) and others deduced from them seem to fit published data reasonably well and in a physically comprehensible way[11,12]. But it is difficult to see how equations of this kind could be deduced without working in pressures.

### 4.4 Reverberation time

Reverberation time is an important factor in defining the acoustic properties of rooms. For a given frequency it is defined as the time for the sound level to drop 60 dB from a steady state after the noise input ceases. "It is assumed ...", to quote Larsen[13], "that the decay rate is exponential and therefore manifests itself as a straight line when the sound pressure level is represented on a logarithmic scale."

*That is, it we start from a level* $D_1$ *dB* $= 20\log(p_1/p_r)$, it is assumed that the level at time $t$ is $D$ dB $= 20\log(p/p_r)$

where $$p = p_1 e^{-(t/T)} \quad (18)$$

$p_r$ being the reference pressure and $T$ a time constant.

Converting to dB, equation (18) becomes

$$D = D_1 - 20(t/T)\log e = D_1 - 8.686(t/T) \quad . \quad (19)$$

This equation clearly represents a straight line of dB against a linear time scale.

The reverberation time, $T_r$, is that time at which $D_1 - D = 60$, that is

$$T_r = 3T/\log e = 6.908\,T \quad . \quad (20)$$

If these equations represent conditions accurately, there would be no objection to measuring the reverberation time by taking the slope of the line at any time. But, in fact, decay curves generally exhibit some curvature, the slope decreasing as the level falls. Larsen[13] suggests that this curvature is

due to the involvement of different Eigen modes, and obtained quite good correlation between theory and his experimental results except in one case: decay at 63 Hz. He explained this by saying that possibly the frequency spectrum was not flat at 63 Hz or "As the level at 63 Hz was also low, the background noise could have had some influence on the measured curve."

Indeed it could! Equation (18) is only an approximation, and only holds as long as the levels are well above the background level, the true equation must be that given by equation (16)

$$p - p_2 = (p_1 - p_2)e^{-(t/T)}$$

where $D_2 = 20 \log(p_2/p_r)$ is the level of the background noise.

If the initial level is $D'$ dB above the background level, that is

$$D' = D_1 - D_2$$

we may re-write this equation in terms of dB and $T_r$ as

$$D = 20 \log\left\{1 + \left[10^{D'/20} - 1\right]10^{-(3t/T_r)}\right\} . \qquad (21)$$

Fig 5a shows $D$ plotted against $(t/T_r)$ for $D' = 70$ . It will be seen that the curve is very nearly linear, at least to the eye, until $D$ becomes small, when the slope decreases and apparent reverberation times become very large.

When decay curves exhibit noticeable curvature, or when the total drop is insufficient to measure the time taken to decay 60 dB, or even 30 dB, various other methods of estimating the reverberation time have been tried. Larsen, for example, compared the estimates made from the slopes of the curves and those made from best linear fits.

It is easy to show by differentiating equation (21), that the slope of the decay curve shown in Fig 4a at a level $D$ dB above the ultimate level is

$$-60(1 - 10^{-D/20})/T_r$$

so that the reverberation time $T_r'$ estimated from the tangent is given by

$$T_r'/T_r = 1/(1 - 10^{-D/20}) . \qquad (22)$$

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Curve 1 of Fig 5b shows the percentage increase of $T'_r$ over $T_r$ plotted against D . It will be seen that the error will be less than 5 per cent so long as D exceeds about 26 dB, but for lower values of D increases very greatly.

The 'best fit' method over a drop from D dB to d dB is likely to give a slope approximating to the line joining those extreme points. It can be shown that the estimate of $T_r$ from such a line will be given by

$$T'_r/T_r = \left[20/(D - d)\right] \log\left[(10^{D/20} - 1)/(10^{d/20} - 1)\right] . \qquad (23)$$

Curves 2 and 3 of Fig 5b show the percentage increase of $T'_r$ over $T_r$ for drops to 10 dB and 5 dB above the ultimate level. It will be seen that the errors always exceed 5 per cent even for initial levels as high as 70 dB. The moral of this analysis is that plots of level against time will not give an accurate estimate of reverberation time unless the lowest level used is well above ambient level, probably at least 20 dB. There are other causes of curvature in decay curves, as Larsen has demonstrated[13], this analysis merely explains one simple cause which may sometimes be neglected in the investigation of more complex and abstruse causes.

## 5 CONCLUSION

No general conclusion can be drawn from the somewhat disjointed discussions presented in this Memorandum, except that I personally have grave doubts of the universal utility of the dB in noise work and believe that I have cause for those doubts. I do not expect that my arguments will persuade anyone that noise should never be measured in dB, indeed, I myself do not think that. What I do believe, and what I have tried to express and illustrate in this paper, is that though the dB may be a useful unit in some applications, it is not absolutely necessary, and in many cases, for example when considering noise doses, it complicates a fundamentally simple procedure; and in others such as the study of the dynamics of TTS it prevents the recognition of well-known dynamic forms.

I would conclude with a plea to workers on noise that they should always keep clearly in mind that noise consists of pressure variations and that the dB is merely a method - sometimes convenient, sometimes the reverse - of describing widely varying numbers. If I have persuaded anyone that he should try to write so that those not steeped in dB may understand, I shall feel that my efforts have not been wasted.

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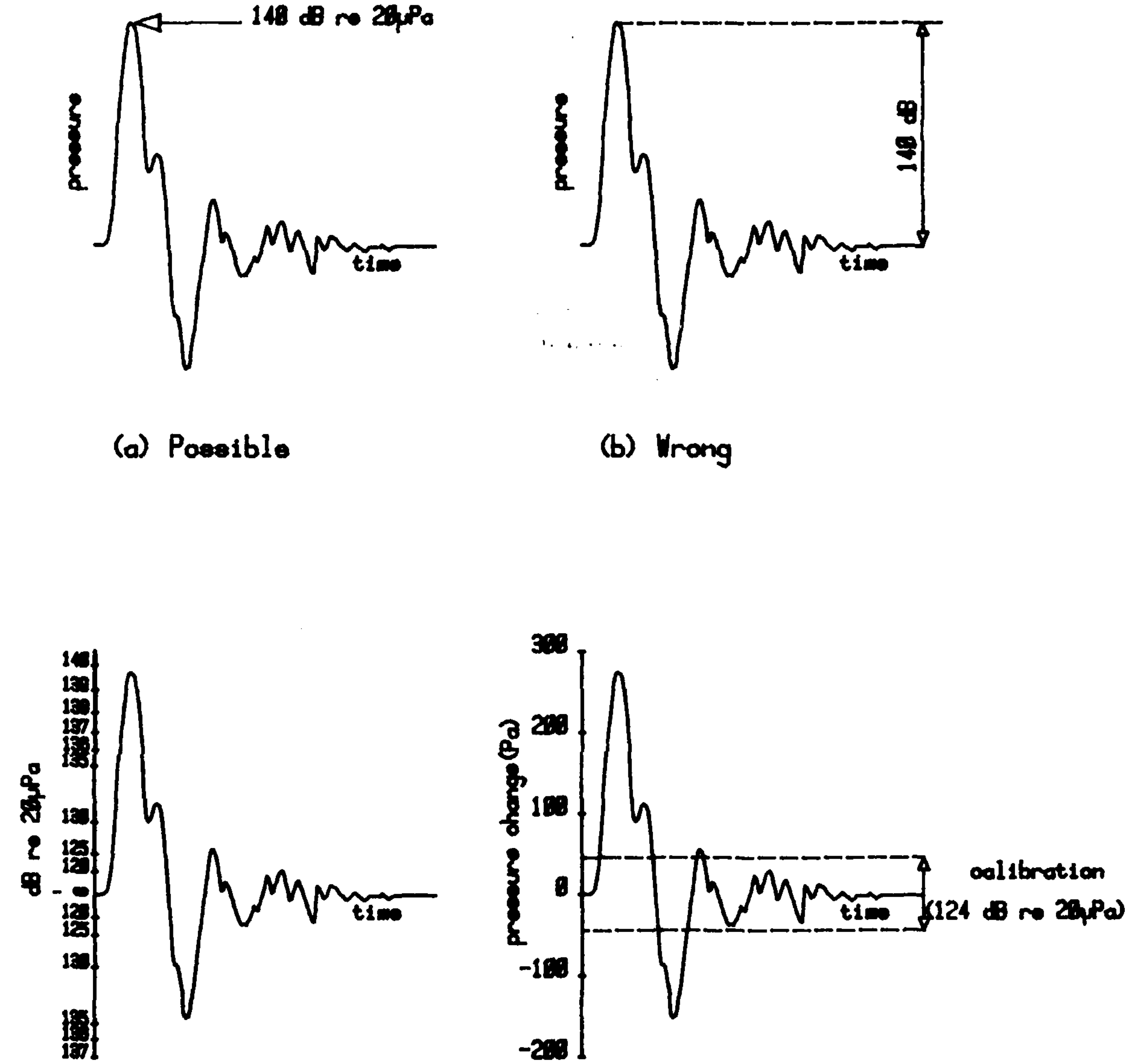


(a) Possible (b) Wrong

(c) Possible (d) Correct

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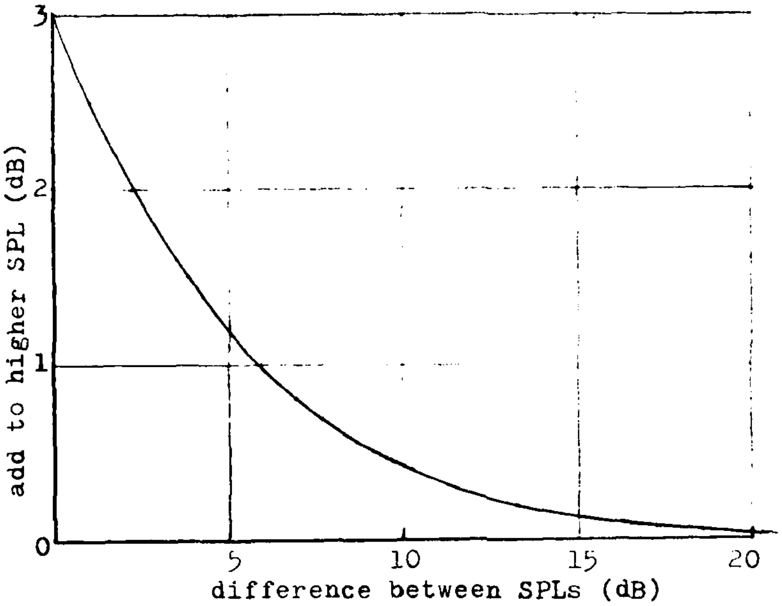
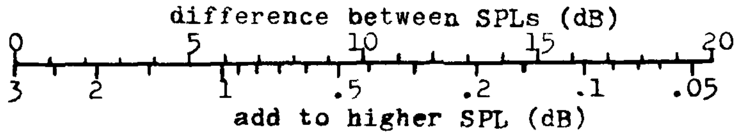
Fig 1 Impulse noise

Fig 2




SPL of noise consisting of two parts x dB and y dB =

$$10 \log(10^{x/10} + 10^{y/10}) = x + 10 \log(1 + 10^{(x-y)/10})$$

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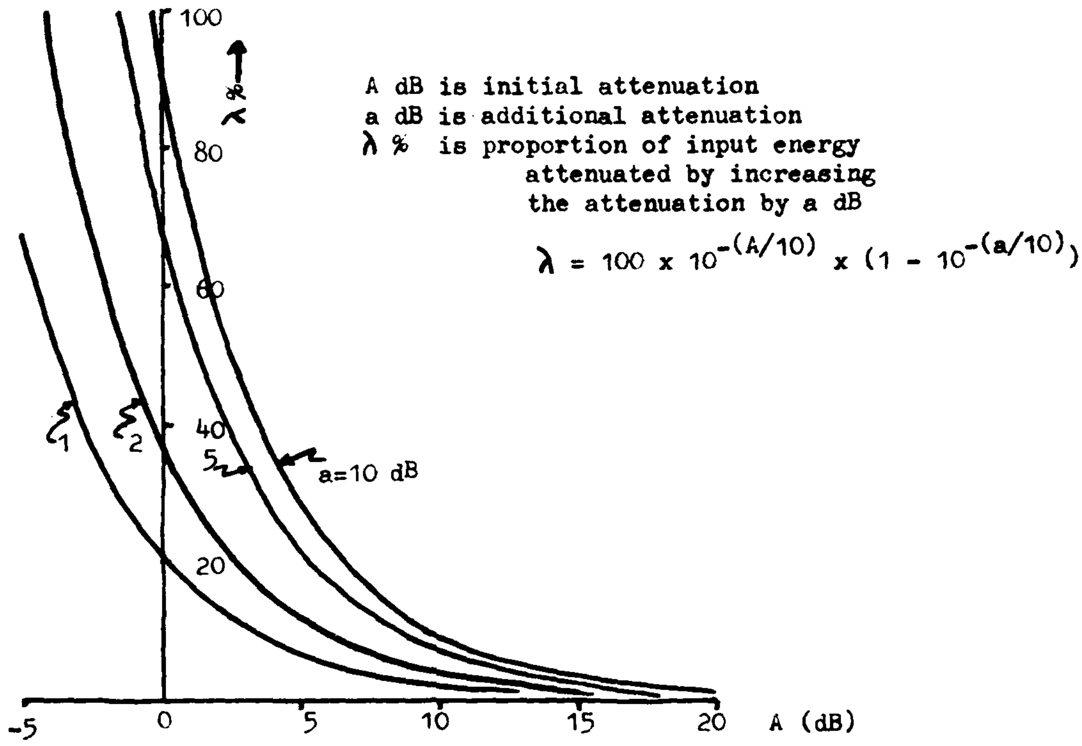
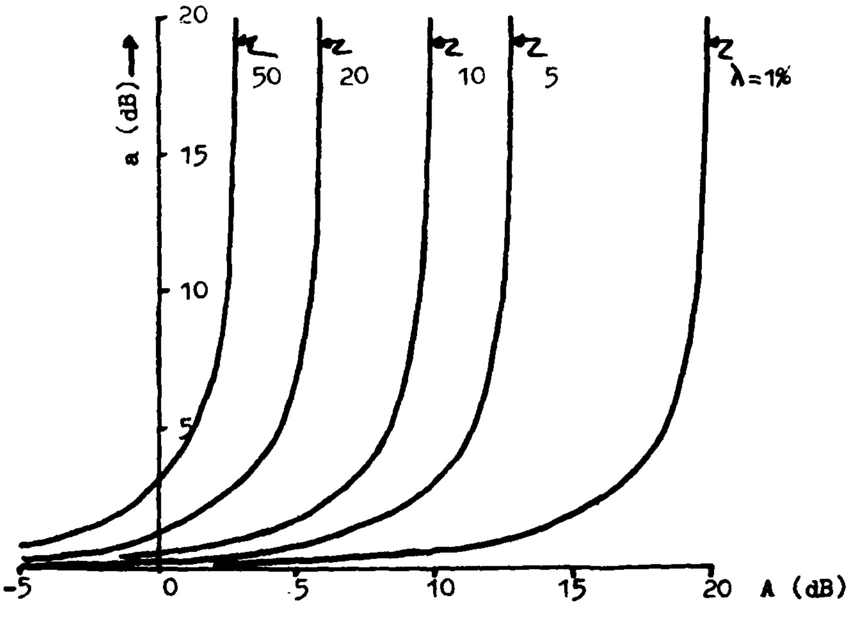
Fig 2 Adding noise levels




(a) Variation of λ with A (a as parameter)




(b) Variation of a with A (λ as parameter)

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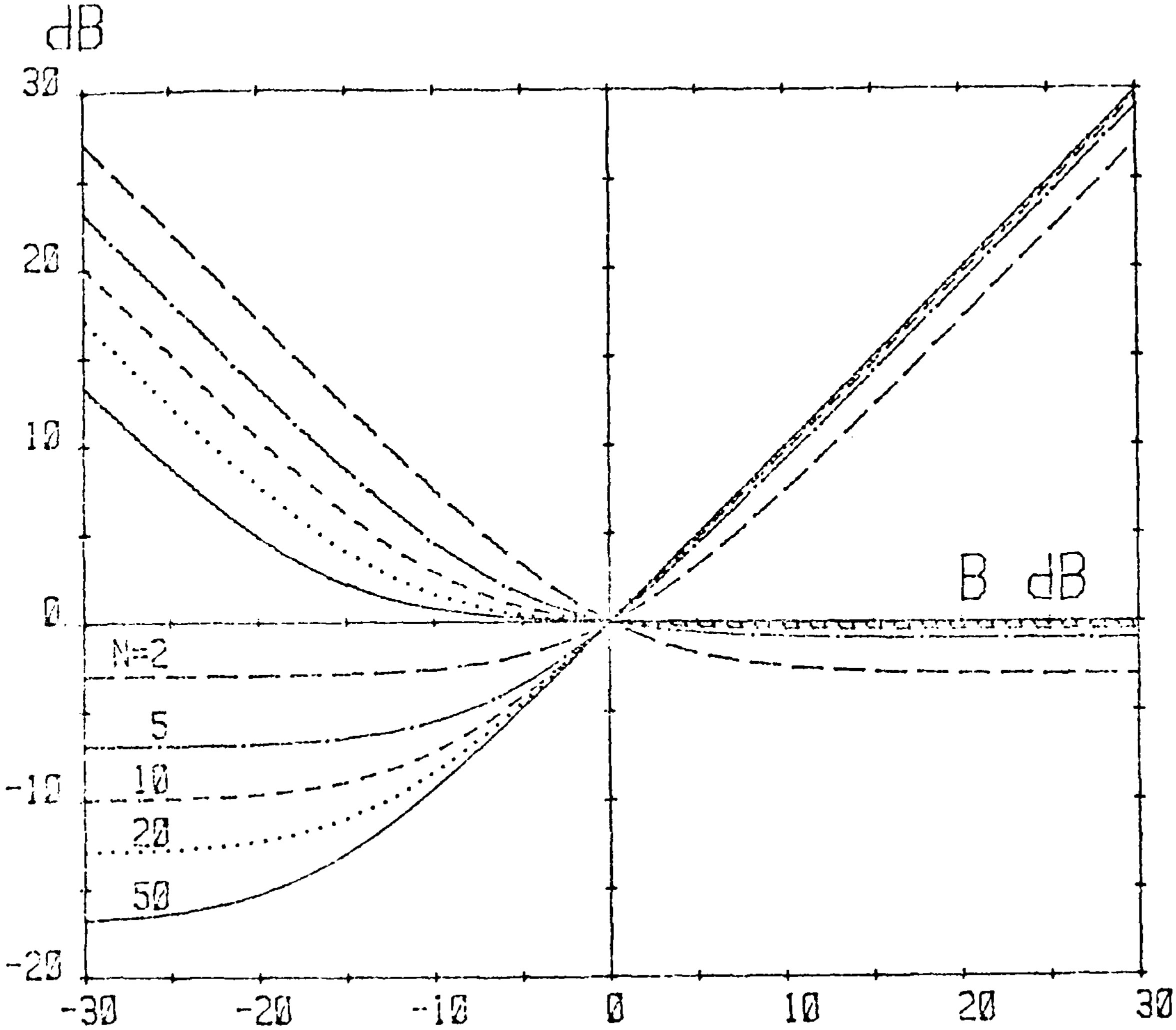
Fig 3 Effect of changes in attenuation on the proportion of energy transmitted

Fig 4



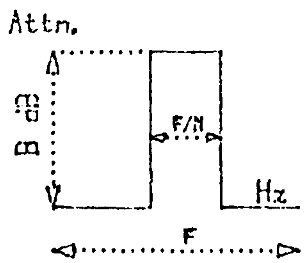
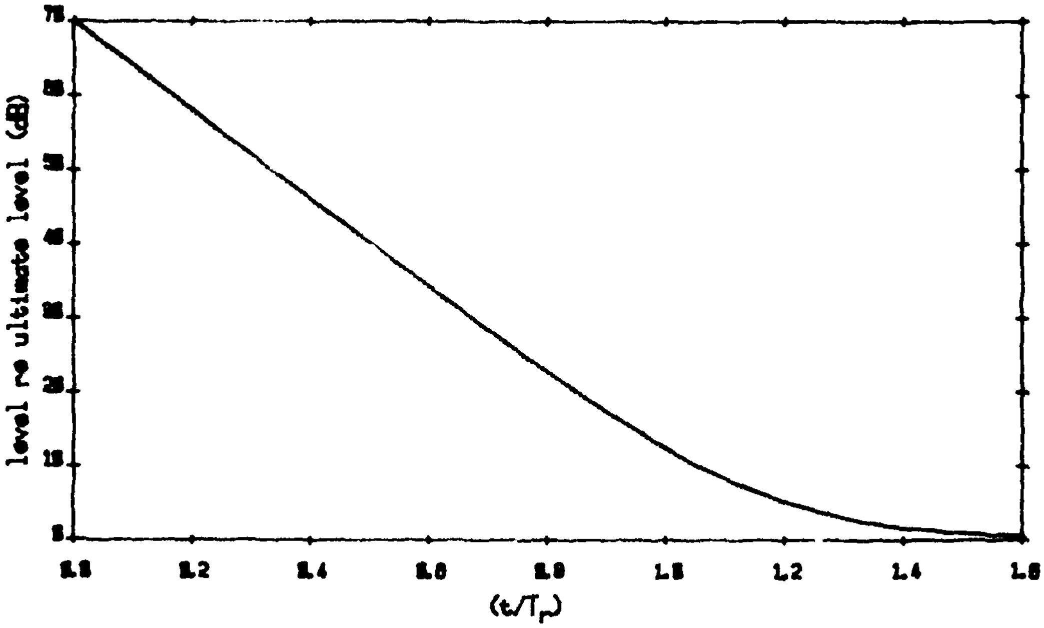
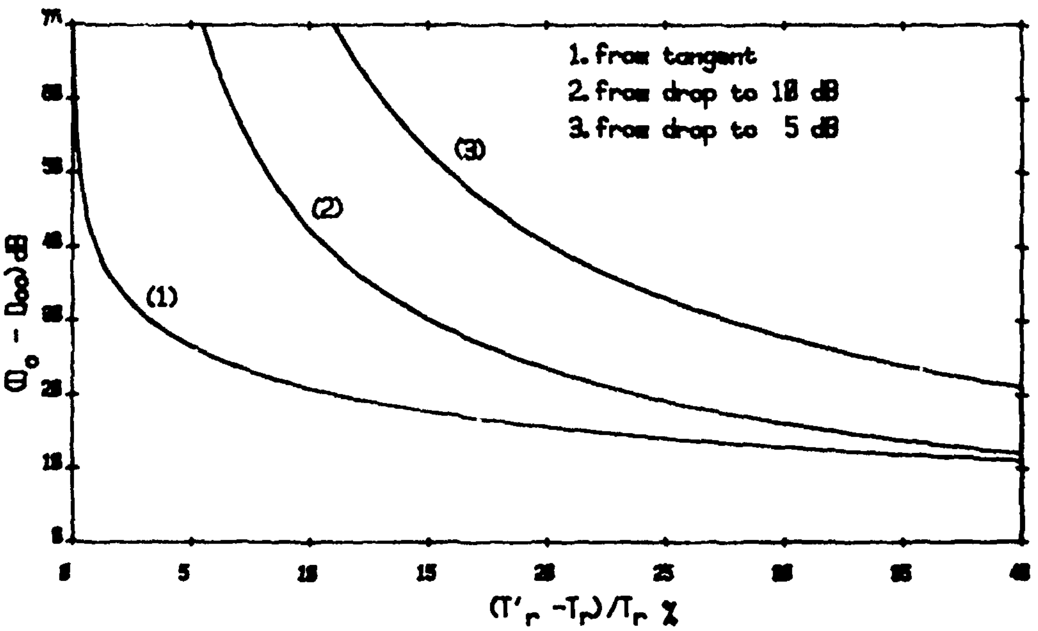




Fig 4 Variation of maximum and minimum attenuation in a band with respect to attenuation of white noise (band contains a step change of attenuation)




a  Noise decay curve




b  Variation of apparent reverberation time with method of measurement

Fig 5a&b  Reverberation time

# REPORT DOCUMENTATION PAGE

Overall security classification of this page

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17. Abstract

This Memorandum presents the author's opinion that the conventional use of the dB in studies of acoustic noise can obscure and complicate [illegible] and lead to the possibility of error because of [illegible] of logarithms and because the order of [illegible] compressed scale. Examples of errors and [illegible] where consideration of noise in terms of the [illegible] lead to better understanding of the processes [illegible] suggested that, although the dB may be a convenient [illegible] emphasis on the physical nature of noise would help [illegible] phenomena and prevent some of the confusions [illegible] with the general public.

DA
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